IN THE SPECIFICATION:

Please replace the paragraph at page 1, line 19 – page 2, line 17 with the following amended paragraph:

Fig. 1 shows the block diagram of a conventional DAC speech synthesizer 100 which includes three basic units, volume control unit 101, signal transform unit 102 and drive unit 103. The volume control unit 101 receives a control signal Vctrl and then generates a control bias Vbias, the signal transform unit 102 receives the control bias Vbias and PCM codes to transform into an analog speech signal Ivo, and the drive unit 103 receives the analog speech signal Ivo and amplifies it to be a current Ispeaker to drive a speaker 104. Fig. 2A is the waveform of a 7-bits sinusoidal PCM signal, Fig. 2B is the waveform of the analog speech signal Ivo after the PCM signal shown in Fig. 2A is processed by the signal transform unit 102 shown ~~is~~ in Fig. 1, and the waveform of the output current signal Ispeaker after the analog speech signal Ivo is amplified by the drive unit 103 is shown in Fig. 2C. As shown in Fig. 2C, when a conventional DAC speech synthesizer transforms a digital speech signal back to an analog signal, the current signal Ispeaker has a zero point about 300 mA, which leads to a more power consumption as shown in the area with dashed lines in Fig. 2C. For applications of

portable electronic products whose power supply is battery, such large power consumption should be avoided. Moreover, to prevent the transistor 105 within the drive unit 103 from being saturated and ~~to~~ resulting in a speech distortion, a bypass resistor 106 is inserted therefor ~~thereof~~, which further results in ~~the~~ speech distortion more seriously.

Please replace the paragraph at page 7, line 13 – page 8, line 6 with the following amended paragraph:

Fig. 6 shows an implemented circuit diagram of the signal transform unit 52, which receives the control bias Vbias and a series of digital speech signal D [0:6], ~~and then~~ to transforms into an analog speech signal Ivo. As shown in the figure, the signal transform unit 52 includes a switched buffer 521 and a switched inverter buffer 522 connected in parallel, and a DAC 523. The switched buffer 521 and inverter buffer 522 receive the lower bits data D [5:0] of the PCM digital speech signal under the control of the MSB D6 in a manner that the switched buffer 521 is enabled to transfer the lower bits data D [5:0] to the DAC 523 when MSB=1; and the switched inverter buffer 522 is enabled to transfer the inverse of the lower bits data D [5:0] to the DAC 523 when MSB=0.

The DAC 523 transforms the lower bits data D [5:0] transmitted from the switched buffer 521 and inverter buffer 522 into the analog speech signal Ivo. As shown in Fig. 7A, a 7-bits sinusoidal PCM digital speech signal has a zero position of 40H, and thus the MSBs of ~~whose~~ those upper and lower half cycles are 1 and 0 respectively. The PCM digital speech signal is therefore transformed by the signal transform unit 52 into the analog speech signal Ivo shown in Fig. 7B.